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If the application is not reading the data as fast as it arrives, the buffer will fill up

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If the space left is very small, the sender can slow down sending until space in the receiver is freed up

А

Initially B has space 100 in its buffer

В

100



A sends 80 bytes

A B

B save the data in its buffer



On the next returning segment, B advertises 20



A now knows it shouldn't send more than 20



Next advertisement would be 0

A B 40

Until B reads some of the data



And can advertise the space

Thus B can tell A to slow down or speed up as appropriate to its remaining buffer space

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The other flow control mechanism to deal with varying conditions in the network comes later

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Urgent data includes things like interrupts that need to be processed before any other data that is buffered



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The OS interrupt code would then read through the urgent data block and act appropriately on what it finds there

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This flag says send the buffered data to the application, don't wait

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The idea here is that there is no point for the receiver waiting for more data, as there is no more to send right now



After the fixed header there are the options, including *window scale* and *maximum segment size*



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After the options header is the data, which can be empty, e.g. for a pure ACK
TCP Options are many and varied



Some TCP optional headers

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Maximum segment size (MSS) specifies how large a segment we can cope with: the headers are not included in count

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The MSS is usually communicated in the option header in the setup of the TCP connection, and is typically set to avoid fragmentation

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A large window is very important is modern fast networks to get the most out of the available bandwidth: we don't want the client to have to keep stopping to wait for the server

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Exercise Go back and re-read the section on advertised windows

Timestamp (TS val) puts the time of day into the segment header, allowing accurate measurement of the *round trip time* (RTT) of a segment and its ACK. Useful for computing retransmission times (see later)

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Timestamp Echo Reply (TS ECR) in an ACK segment is the timestamp being returned to the sender so it can compute the RTT

Selective acknowledgement (SACK) is an extension of the ACK mechanism that allows more flexible ways of acknowledging segments. SACK is negotiated in the connection setup with a *SACK Permitted* option

Several options are only allowed in the first segment of a new connection, e.g., Window scale, MSS and SACK Permitted

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This is because some things, e.g., buffer space, need to be set up before a connection and varying them mid-connection is difficult or makes little sense

TCP is *connection oriented*, meaning a connection is set up between source and destination, and all packets that flow within this connection are related, through the sequence numbers, and they all use the same state, such as advertised window

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It is important to realise that this is a connection in the *transport layer*

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Thus TCP connection has a weak version of sessions: though no further session mechanism is provided, e.g., no session resumption



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Of course, both connection oriented and connectionless protocols are useful in the right circumstances

TCP

Setup and Teardown

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Fortunately for the application programmer, all this detail is taken care of by the TCP layer software in the operating system: though it does have occasional repercussions in the application if the connection needs to outlive the application for a while



Before TCP can send data, it exchanges some packets with the setup information
Setup and Teardown



Three segments are needed to exchange the information needed to make a new connection;

Setup and Teardown



The initiator, the *client*, sends a segment with the SYN flag set and its *initial sequence number* (ISN), *n*, is randomly generated;

Setup and Teardown



The receiver, the *server*, replies with another SYN segment containing its own ISN, *m*;

Setup and Teardown



It also ACKs the client's ISN with n + 1, the sequence number of the next byte it expects from the client;

Setup and Teardown



The initial SYN can be lost just like any other segment, so we need to ACK it independently of the first data byte, which comes later;

Setup and Teardown



The client ACKs the server's ISN with m + 1

TCP Setup and Teardown

This is called a three way handshake



These segments contain no user data: they are overhead in setting up the connection



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Overhead in time and overhead in packets on the network



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After the handshake we can start sending data

The client (first one to initiate) is said to do an *active open*, while the server does a *passive open*

Setup and Teardown



Establishment TCP simultaneous open

It is possible (but rare) for *both* hosts to do an active open, where the SYNs cross each other in flight

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Matching TCP port numbers will identify when this happens

This is defined to produce one new connection, not two

Setup and Teardown



Closing a connection takes up to four segments;

Setup and Teardown



TCP is full duplex, and a connection in one direction may be closed independently of the other;

Setup and Teardown



The FIN flag is set to indicate a *half close*: this indicates no more data will be sent from this end;

Setup and Teardown



We can still *receive* data at this end;

Setup and Teardown



TCP teardown

The FIN is ACKed;

Setup and Teardown



When the other end wants to close, it sends a FIN and gets an appropriate ACK;

Setup and Teardown



Note there may still be data (and the corresponding returning ACKs) flowing from the server to the client before the server decides to close;

Setup and Teardown



The first close is called an *active close*;

Setup and Teardown



TCP teardown

The other end does a *passive close*

Setup and Teardown



Either end can initiate the active close; it does not need to be the host that did the active open

Setup and Teardown



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Setup and Teardown



The passive close FIN can be piggybacked on the ACK: this then takes only three segments

TCP Setup and Teardown



Termination

Simultaneous active close

There can (rarely) be a simultaneous active close: this takes four segments again

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For example, if a server crashes and reboots while the client is still sending the server will not know what to do with the segments it is receiving; so it replies with a RST

When a host gets a RST it ends the connection immediately, discarding all state and buffered segments



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Often seen by the application as a "connection reset by peer" message

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RSTs are not ACKed: the connection ends right here

Exercise Think about the security aspects of this: a third party can inject a RST segment into a connection to kill it